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10/757,994	01/16/2004	Changkyu Choi	1793.1172	5012

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EXAMINER	
PAUL, DISLER	

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Please find below and/or attached an Office communication concerning this application or proceeding.

The time period for reply, if any, is set in the attached communication.

Office Action Summary	Application No.	Applicant(s)	
	10/757,994	CHOI ET AL.	
	Examiner	Art Unit	
	Disler Paul	2615	

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE 3 MONTH(S) OR THIRTY (30) DAYS, WHICHEVER IS LONGER, FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☐ Responsive to communication(s) filed on _____.

2a)☐ This action is **FINAL**. 2b)☒ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☒ Claim(s) 1-30 is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) 1-30 is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☐ The specification is objected to by the Examiner.

10)☐ The drawing(s) filed on _____ is/are: a)☐ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

Priority under 35 U.S.C. § 119

12)☒ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☐ All b)☐ Some * c)☐ None of:

1.☒ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1) ☒ Notice of References Cited (PTO-892)

2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)

3) ☒ Information Disclosure Statement(s) (PTO/SB/08) Paper No(s)/Mail Date *02/02/06 and 5/10/04*.

4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____ .

5) ☐ Notice of Informal Patent Application

6) ☐ Other: _____.

DETAILED ACTION

Claim Rejections - 35 USC § 102

1. The following is a quotation of the appropriate paragraphs of 35 U.S.C. 102 that form the basis for the rejections under this section made in this Office action:

> A person shall be entitled to a patent unless –
>
> (b) the invention was patented or described in a printed publication in this or a foreign country or in public use or on sale in this country, more than one year prior to the date of application for patent in the United States.

2. Claims 1-9; 25-27 are rejected under 35 U.S.C. 102(b) as being anticipated by Hoshuyama ("6,449,586 B1").

Re claim 1, Hoshuyama disclose of an adaptive beamforming method (fig.1-10), comprising: compensating for time delays of M noise-containing speech signals input via a microphone array having M microphones, wherein M is an integer greater than or equal to 2, and generating a sum signal of the M compensated noise-containing speech signals (fig.1/(2,4) plurality of delayed noise-speech signals generated to (20); col.2 line 16-35); and extracting pure noise components from the M compensated noise-containing speech signals using M adaptive blocking filters that are connected to M adaptive canceling filters in a feedback structure (fig.1 (20) with blocking feedback filters to extract noise via (6); col.4 line 1-4; and connect to (7)); and extracting pure speech components from the sum signal using the M adaptive canceling filters that are connected to the M

adaptive blocking filters in the feedback structure (fig.1 (30) with canceling filters in feedback; col.5 line 25-40).

Re claim 2, the method of claim 1, wherein the extracting pure noise components comprises: filtering a noise-removed sum signal through the M adaptive blocking filters and subtracting signals output from the M adaptive blocking filters from the M compensated noise-containing speech signals to output M noise signals (fig.1/20 with blocking filters to be subtracted (6)) and further filtering the M noise signals through the M adaptive canceling filters and subtracting signals output from the M adaptive canceling filters from the sum Signal (fig.1(12,9) sum to the subtractor; col.5 line 30-40) and inputting M subtraction results to the M adaptive blocking filters as the noise-removed sum signal and adding the M subtraction results (fig.1(5,6) wt feedback as result back to filter).

Re claim 3, the method of claim 1, wherein the extracting pure noise signals comprises: filtering a noise-removed sum signal through the M adaptive blocking filters (fig.1 (3,5) noise/target signals to filters); subtracting signals output from the M adaptive blocking filters from the M compensated noise-containing speech signals to output M noise signals (fig.1 (6)); filtering the M noise signals through the M adaptive canceling filters (fig.1 (7)); adding signals output from the M adaptive canceling filters and outputting an adaptive canceling filter sum signal (fig.1 (12)); and subtracting the

adaptive canceling filter sum signal from the sum signal (fig.1 (9)) and inputting M subtraction results to the M adaptive blocking filters as the noise-removed sum signal(fig.1(5,6) wt feedback as result back to filter).

Re claim 4, the method of claim 2, wherein the M adaptive blocking filters and the M adaptive canceling filters are finite impulse response filters (col.2 line 48-52).

Re claim 5, the method of claim 4, wherein coefficients of the M adaptive blocking filters and the M adaptive canceling filters are updated by an information maximization algorithm (col.3 line 30-67; col. 5 line 10-37).

Re claim 6-7, have been analyzed and rejected with respect to claim 4-5 respectively above.

Re claim 8, Hoshuyama disclose of an adaptive beamforming apparatus (fig.1-10), comprising: a fixed beamformer that compensates for time delays of M noise-containing speech signals input via a microphone array having M microphones, wherein M is an integer greater than or equal to 2, and generates a sum signal of the M compensated noise-containing speech signals (fig.1(20); col.2 line 1—35); and a multi-channel signal separator that extracts pure noise components

from the M compensated noise-containing speech signals using M adaptive blocking filters that are connected to M adaptive canceling filters in a feedback structure and extracts pure speech components from the sum signal using the M adaptive canceling filters that are connected to the M adaptive blocking filters in the feedback structure (fig.1 (20,30) blocking and adaptive filters with feedback to extrack noise and speech signals respectively by each filter and see claim 1 explanation).

Re claim 9, the apparatus of claim 8, wherein the fixed beamformer comprises: a delay unit that delays the M noise-containing speech signals by the time delays (fig.1 (4m)/input signals to be delayed; col.2 line 40-47); and a first adder that adds the M noise-containing speech signals delayed by the delay (fig.1(6)), while Hoshuyama disclose of the above, He is silent in regard of the time delay estimator that calculate the time delays of the M noise-containing speech signals, however, with the above disclose information of time delays being determined, it is inherent that there must exist such a time delay estimator.

Re claim 25, Hoshuyama disclose of the method of removing noise from time delayed signals subject to noise, comprising: receiving signals having noise components; delaying the received signals having the noise components by a predetermined period of time to generate delayed received signals; adding the delayed received signals to

generate a combination received signal (fig.1 (3); col.3 line 5-13/beamformers delayed as groups); generating separate clean signals without noise components using adaptive feedback filtering based on the delayed received signals, the combination received signal, and the separate clean signals (fig.1 (5)/ with delay and total and feedback for clean signals); and generating a clean signal without noise components using the separate clean signals (fig.1 (5,7) clean signals is generated).

Re claim 26, the method of claim 25, wherein using adaptive feedback filtering comprises: generating separate clean signals without noise components by subtracting noise components, output from adaptive canceling filters having predetermined coefficients, from the combination received signal (col.4 line 45-67); generating separate noise signals by subtracting signals output from adaptive blocking filters having predetermined coefficients, which receive the separate clean signals, from the delayed received signals (fig.1 (4,5); col.3 line 30-50).

Re claim 27, the method of claim 26, wherein generating the clean signal without noise components comprises adding the separate clean signals (fig.1(5) with feedback clean signals updated).

Claim Rejections - 35 USC § 103

3. The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

4. Claims 10-15; 17; 21-24 are rejected under 35 U.S.C. 103(a) as being unpatentable over Hoshuyama ("6,449,586 B1") and further in view of Bhadkamkar et al. ("60002,776).

Re claim 10, the apparatus of claim 8, wherein the multi-channel signal separator comprises: a first filter that filters a noise-removed sum signal through the M adaptive blocking filters; a first subtractor that subtracts signals output from the M adaptive blocking filters from the M compensated noise-containing speech signals using M subtractors (fig.1 (20) with filters (5) and subtractor); a second filter that filters M subtraction results of the first subtractor through the M adaptive canceling filters (fig.1 (30/7)). While, Hoshuyama disclose of the above, He fail to disclose of the second subtractor that subtracts signals output from the M adaptive canceling filters from the sum signal using M subtractors. But, Bhadkamkar et al. disclose of an audio processing system wherein the second subtractor that subtracts signals output from the M adaptive canceling filters from the sum signal using the M subtractor (fig.3,5,7 (56,54)

in addition to first filter and substractor for the plurality of input microphones) for the purpose of leaving signal component only from the original source B. thus, taking the combined teaching of Hoshuyama and Bhadkamkar et al. as a whole, it would have been obvious for one of the ordinary skill in the art to modify Hoshuyama by incorporating the second subtractor that subtracts signals output from the M adaptive canceling filters from the sum signal using the M subtractor for the purpose of leaving signal component only from the original source B.

The combined teaching of Hoshuyama and Bhadkamkar et al. as a whole, further teach of the inputs M subtraction results to the M adaptive blocking filters as the noise-removed sum signal (Badkamkar,fig.3,5,7(52)/ subtraction result into the adaptive filters inputted) ; The combined teaching of Hoshuyama and Bhadkamkar et al. as a whole, would have incorporate the second adder that adds signals output from the M subtractors of the second subtractor (Hoshuyama, fig.1 (9)).

Re claim 11, the apparatus of claim 8, wherein the multi-channel signal separator comprises: a first filter that filters a noise-removed sum signal through the M adaptive blocking filters (fig.1 (20)); a second filter that filters signals output from the M subtractors of the first subtractor through the M adaptive canceling filters (fig.1(30)); a second adder that adds signals output from M

adaptive canceling filters of the second filter (fig.1 (9)); and a second subtractor that subtracts signals output from the second adder from the signals output from the fixed beamformer and inputs M subtraction results to the M adaptive blocking filters as the noise-removed sum signal (fig.1(6)). While Hoshuyama disclose of the above, he fail to disclose of the a first subtractor that subtracts signals output from the M adaptive blocking filters from the M compensated noise-containing speech signals using M subtractors. However, Bhadkamkar et al. disclose of an audio processing system wherein the first subtractor that subtracts signals output from the M adaptive blocking filters from the M compensated noise-containing speech signals using M subtractors (Badkamkar,fig.3,5,7(56,54)/ subtraction result from adaptive filter) for the purpose of leaving signal component only from the original source B. thus, taking the combined teaching of Hoshuyama and Bhadkamkar et al. as a whole, it would have been obvious for one of the ordinary skill in the art to modify Hoshuyama by incorporating the first subtractor that subtracts signals output from the M adaptive blocking filters from the M compensated noise-containing speech signals using M subtractors for the purpose of leaving signal component only from the original source B.

Re claim 12, the apparatus of claim 10, wherein the M adaptive blocking filters and the M adaptive canceling filters are finite impulse response filters (col.2 line 48-52).

Re claim 13, the apparatus of claim 12, wherein coefficients of the M adaptive blocking filters and the M adaptive canceling filters are updated by an information maximization algorithm (col.3 line 30-67; col. 5 line 10-37).

Re claims 14-15 have been analyzed and rejected with respect to claims 12-13 respectively.

Re claim 17, the apparatus of claim 16, wherein the feedback structure of the signal separator comprises: a plurality of first subtractors that receive the delayed received signals and subtract corresponding signals from the plurality of adaptive blocking filters to output separate noise component signals (fig.1 (6m)); and the plurality of adaptive canceling filters receive the corresponding separate noise component signals as inputs (fig.1(30) input adaptive filters). While, Hoshuyama disclose of the above, He fail to further disclose of the plurality of second subtractors that receive the combination received signal and subtract corresponding signals from the plurality of adaptive canceling filters to output separate clean signals without noise components. But, Bhadkamkar et al. disclose of an audio processing system wherein the second subtractors that receive the combination received signal and subtract corresponding signals from the adaptive canceling filters to output separate clean signals without noise components ("fig.3,5 /wt subtractor (56))for the purpose

of leaving signal component only from the original source B. thus taking the combined teaching of Hoshuyama and Bhadkamkar et al. as a whole, it would have been obvious for one of the ordinary skill in the art to modify Hoshuyama by incorporating the plurality of second subtractors that receive the combination received signal and subtract corresponding signals from the plurality of adaptive canceling filters to output separate clean signals without noise components for the purpose of leaving signal component only from the original source B.

The combined teaching of Hoshuyama and Bhadkamkar et al. as a whole, would have incorporate further of the plurality of adaptive blocking filters receive the corresponding separate clean signals without noise components as inputs (Bhadkambar,fig.3(54)).

Re claim 21, the apparatus of claim 16, wherein the feedback structure of the signal separator comprises: a plurality of first subtractors that receive the delayed received signals and subtract corresponding signals from the plurality of adaptive blocking filters (fig.1(6)), and the plurality of first subtractors outputs signals to the plurality of adaptive canceling filters (fig.1 (7)); an adder that adds signals output from the plurality of adaptive canceling filters to output a total noise component signal (fig.1(8)). While Hoshuyama disclose of the above, He fail to further disclose of the second subtractor that receives the combination received signal and subtracts

the total noise component signal to output a clean signal without noise components. But, Bhadkamkar et al. disclose a sound processing wherein the second subtractor that receives the combination received signal and subtracts the total noise component signal to output a clean signal without noise components (fig.5 (56))and fig.3) for the purpose of leaving signal component only from the original source B. thus, taking the combined teaching of Hoshuyama and Bhadkamkar et al. as a whole, It would have been obvious for one of the ordinary skill in the art to modify Hoshuyama by incorporating the second subtractor that receives the combination received signal and subtracts the total noise component signal to output a clean signal without noise components for the purpose of leaving signal component only from the original source B.

The combined teaching of Hoshuyama and Bhadkamkar et al. as a whole, further teach of the plurality of adaptive blocking filters receive the clean signal without noise components as an input and the adaptive blocking filters generate signals corresponding to a portion of the clean signal without noise components of the delayed received signals to the plurality of first subtractors (Bhadkambar,fig.3(54)).

Re claim 22, the apparatus of claim 17, wherein the adaptive blocking filters and the adaptive canceling filters are finite impulse response filters (col.2 line 48-52).

Re claim 23, the apparatus of claim 18, wherein the blocking coefficients and the canceling coefficients are updated automatically by an information maximization algorithm (col.3 line 30-67; col. 5 line 10-37 & see claims 1 for explanation).

Re claim 24, the apparatus of claim 19, wherein a number of taps necessary to implement the feedback structure is optimized (col.3 line 12-13; line 30-45).

5. Claim 16l 18-20; 28-30 are rejected under 35 U.S.C. 103(a) as being unpatentable over Hoshuyama ("6449,586 B1).

Re claim 16, Hoshuyama disclose of the adaptive beamforming apparatus (fig.1-10), comprising: a receiver that receives signals including noise components, delays the received signals by a calculated time to provide delayed received signals (fig.1 (20,40)/delay signals), and adds the delayed received signals to provide a combination received signal (fig.1 (3); col.3 line 5-13/beamformers delayed as groups); a signal separator that generates a clean signal without noise components based on adaptively filtering the delayed received signals and the combination received signal by a plurality of adaptive blocking filters having blocking coefficients (fig.1 (20) plurality of blocking filters with (3,4) group and delayed

signals; col.3) and a plurality of adaptive canceling filters having canceling coefficients connected in a feedback structure, wherein the blocking coefficients and the canceling coefficients are updated during operation of the signal separator(co.4 line 5-60;col.3/ fig.1 filters wt feedback). While, Hoshuyama disclose of the above limitation, he fail to further disclose of the specific of the coefficients are updating automatically, However, official notice is taken that this limitation of automatically updating is commonly known in the art, thus it would have been obvious for one of the ordinary skill in the art to have modified Hoshuyama by incorporating the limitation of automatically updating for purpose of optimizing the system.

Re claim 18, the apparatus of claim 17, wherein the adaptive blocking filters and the adaptive canceling filters are finite impulse response filters (col.2 line 48-52).

Re claim 19, the apparatus of claim 18, wherein the blocking coefficients and the canceling coefficients are updated automatically by an information maximization algorithm (col.3 line 30-67; col. 5 line 10-37).

Re claim 20, the apparatus of claim 19, wherein a number of taps necessary to implement the feedback structure is optimized (col.3 line 12-13; line 30-45).

Re claim 29, the method of claim 26, further comprising: updating the coefficients of the adaptive canceling filters and the adaptive blocking filters automatically by an information maximization algorithm (col.3 line 30-67; col. 5 line 10-37 and see claim 16 for automatically explanation).

Re claim 30, the method of claim 26, further comprising: updating the coefficients of the adaptive canceling filters and the adaptive blocking filters automatically by one of a least square algorithm and a normalized least square algorithm (col.3 line 34-45

Re claim 28 has been analyzed and rejected with respect to claim 29 above.

Conclusion

Any inquiry concerning this communication or earlier communications from the examiner should be directed to Disler Paul whose telephone number is 571-2701187. The examiner can normally be reached on 7:30-5:00.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Chin Vivian can be reached on 571-272-7848. The fax phone number for the organization where this application or proceeding is assigned is 571-273-8300.

Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Status information for published applications may be obtained from either Private PAIR or Public PAIR. Status information for unpublished applications is available through Private PAIR only. For more information about the PAIR system, see http://pair-direct.uspto.gov. Should you have questions on access to the Private PAIR system, contact the Electronic Business Center (EBC) at 866-217-9197 (toll-free). If you would like assistance from a USPTO Customer Service Representative or access to the automated information system, call 800-786-9199 (IN USA OR CANADA) or 571-272-1000.

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